UNITED STATES PATENT AND TRADEMARK OFFICE

UNITED STATES DEPARTMENT OF COMMERCE
United States Patent and Trademark Office
Address: COMMISSIONER FOR PATENTS
P.O. Box 1450
Alexandria, Virginia 22313-1450
www.uspto.gov

APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
09/849,719	05/04/2001	Kenneth H.P. Chang	SSI001US	9646

27906 7590 07/28/2004

PATENT LAW OFFICES OF DAVID MILLERS
6560 ASHFIELD COURT
SAN JOSE, CA 95120

EXAMINER	
VO, HUYEN X	
ART UNIT	PAPER NUMBER
2655	4

DATE MAILED: 07/28/2004

Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	**Application No.** 09/849,719	**Applicant(s)** CHANG, KENNETH H.P.	
	Examiner Huyen Vo	**Art Unit** 2655	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE _3_ MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on _04 May 2001_.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) _1-36_ is/are pending in the application.

4a) Of the above claim(s) _26-36_ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) _1-25_ is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on _04 May 2001_ is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☒ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date _2-3_.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application (PTO-152)

6) ☐ Other: _____.

DETAILED ACTION

Election/Restrictions

1. Restriction to one of the following invention is required under 35 U.S.C. 121:

Group I, claims 1-25 drawn to an apparatus and method for processing and presenting multi-channel audio information, which is classified under class 704, subclass 500.

Group II, claims 26-36 drawn to a method for controlling display of web pages, and a method for authoring a presentation for playback on a computing system, which are classified under class 345, subclass 716.

Inventions I and II are related as subcombinations disclosed as usable together in a single combination. The subcombinations are distinct from each other if they are shown to be separately usable. In the instant case, invention I has separate utility such as with a home stereo system in decoding audio from hard copy media, rather than with a system processing, downloaded web pages as in invention II. See MPEP § 806.05(d).

Because these inventions are distinct for the reasons given above and have acquired a separate status in the art as shown by their different classification, restriction for examination purposes as indicated is proper.

During a telephone conversation with Mr. David Millers on 7/8/2004, an election was made without traverse to prosecute the invention 1, claims 1-25. Affirmation of this election must be made by applicant in replying to this Office

action. Claims 26-36 are withdrawn from further consideration by the examiner, 37 CFR 1.142(b), as being drawn to a non-elected invention.

Claim Rejections - 35 USC § 102

2. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (b) the invention was patented or described in a printed publication in this or a foreign country or in public use or on sale in this country, more than one year prior to the date of application for patent in the United States.

3. Claims 1-4, 6-7, 10, 12, and 14-18 are rejected under 35 U.S.C. 102(b) as being anticipated by Smyth et al. (US Patent No. 5974380).

4. Regarding claim 1, Smyth et al. disclose an apparatus containing a data structure representing a presentation, the data structure comprising:

a first audio channel representing an audio portion of the presentation after time scaling by a first time scale factor (*col. 14, ln. 14-33*); and

a second audio channel representing the audio portion after time scaling by a second time scale factor that differs from the first time scale factor (*col. 14, ln. 14-33*).

5. Regarding claim 12, Smyth et al. disclose an apparatus containing a data structure representing an audio presentation, the data structure comprising a

plurality of audio channels representing the audio presentation after time scaling, wherein:

each audio channel has a corresponding time scale factor and includes a plurality of audio frames (*col. 14, ln. 14-33*); and

each audio frame has a frame index that uniquely distinguishes the audio frame from other audio frames in the same channel and identifies the audio frame as corresponding to specific audio frames in other audio channels (*Data Stream Format section on col. 33, ln. 39 to col. 39, ln. 67 or referring to figure 25, well known to a person of ordinary skill in the art*).

6. Regarding claim 14, Smyth et al. disclose a method for encoding audio data, comprising:

performing a plurality of time scaling processes on the audio data to generate a plurality of time-scaled audio data sets, each time-scaled audio data set having a different time scale factor (*col. 14, ln. 14-33*); and

generating a data structure containing a plurality of audio channels respectively corresponding to the plurality of time scaling processes, wherein content of each of the audio channels is derived from the time-scaled audio data set resulting from performing the corresponding time scaling process on the audio data (*col. 14, ln. 14-33*).

7. Regarding claims 2-3, Smyth et al. further disclose that the first audio channel comprises plurality of frames (*col. 3, ln. 23-27*); the second audio

channel comprises plurality of frames that are in one-to-one correspondence with the plurality of frames in the first audio channel (*col. 14, ln. 14-33 and referring figures 12a-b, same bandwidth and same number of frames*); and corresponding frames in the first and second audio channels represent the same time interval of the presentation (*referring to figures 12a-e, processing each block (or time interval of data) of data and then represent it to users*), and each frame in the first audio channel is separately compressed using a first compression method (*using the ADPCM coding method col. 3, ln. 23-40*).

8. Regarding claim 4, Smyth et al. further disclose that the data structure further comprises a third audio channel representing the audio presentation after time scaling by the first time scale factor, wherein each frame in the third audio channel is separately compressed using a second compression method (*col. 14, ln. 14-33, the high-frequency portion can be encoding using either ADPCM or VD method and then it is copied to each channel and scaled by the first scale factor in the first channel*).

9. Regarding claims 6-7, Smyth et al. further disclose that the first audio channel comprises plurality of frames, each frame having an index value that identifies a time interval of the audio portion that the frame represents; the second audio channel comprises plurality of frames, each frame in the second channel having an index value that identifies a time interval of the audio portion that the frame represents (*col. 33, ln. 40-67 or referring to figure 25, that is*

interprets as the size or length of the frame), and each frame in the first and second data channels is separately compressed (*col. 14, ln. 14-33*).

10. Regarding claim 10, Smyth et al. further disclose that the apparatus comprises: data storage in which the data structure is stored (*col. 4, ln. 1-11, buffer is used to hold input data for processing, known to one of ordinary skill in the art*); a decoder connected to receive a data stream, the decoder converting the data stream for perceivable presentation (*figure 31*). Smyth et al. do not disclose selection logic coupled to the data storage and capable of selecting a source channel for the data stream from among a set of channels including the first audio channel and the second audio channel (*element 40 in figure 5 demultiplex the audio stream individual audio channels and direct each audio channels to appropriate processing units*).

11. Regarding claim 15, Smyth et al. further disclose that generating the data structure comprises: partitioning each time-scaled audio data set into a plurality of frames (*col. 14, ln. 14-33 and figure 31, when an audio signal is demultiplexed and decompressed for playback, the decoding system processes the encoded data stream frames by frames*); separately compressing each frame to produce compressed frames (*referring to figures 4a and 5*); and collecting the compressed frames into the plurality of audio channels, each audio channel having a corresponding one of the different time scale factors (*the output 16 in figure 5*).

12. Regarding claims 16-18, Smyth et al. further disclose that all frames resulting from the partitioning correspond to the same amount of time in the audio data (*col. 14, ln. 14-33 and referring figures 12a-b, same bandwidth or time and same number of frames*); separately compressing each frame comprises applying a plurality of different compression processes to generate a plurality of compressed frames from each frame (*col. 14, ln. 14-33*); and collecting the compressed frames produces audio channels such that in each audio channel, all compressed frames in the audio channel have the same time scale and compression process (*referring to figures 25 and 31*).

13. Claims 19, 22, and 23 are rejected under 35 U.S.C. 102(b) as being anticipated by Ware (US Patent No. 5664044).

14. Regarding claim 19, Ware discloses a method for playing a presentation, comprising:

loading a first frame from a source into a player via a network, the first frame representing a first portion of the presentation after scaling by a first time-scaling factor, wherein the first audio frame has a first channel index value that identifies the first audio frame as being scaled by the first time scaling factor (*col. 5, ln. 55 to col. 6, ln. 30, or referring to figure 1*);

playing the first portion of the presentation based on data from the first audio frame (*col. 6, ln. 1-30*); receiving a request to change playing from the first time scaling factor to a second time scaling factor (*col. 6, ln. 31-53*);

requesting from the source a second audio frame that has a second channel index value that identifies the second frame as being scaled by the second time-scaling factor (*referring to figure 1*); and playing the second frame after the first to provide a real-time change in the time-scale of the presentation (*the operation of figure 1 is a continuous process*).

15. Regarding claim 22, Ware further discloses that the channel index values of frames further indicate respective compression processes for the frames, and wherein the method further comprises: determining available bandwidth on the network (*col. 6, ln. 31-67, by calculating flowing conditions, which is related to how fast data are sent*); and selecting the second channel index value from a plurality of channel index values that identify the second time scaling factor, wherein the second channel index indicates a compression process provides highest audio quality at the available bandwidth (*col. 6, ln. 54 to col. 7, ln. 57, a process of adjusting the scale factor to achieve optimum*).

16. Regarding claim 23, Ware further disclose that the channel index values of frames further indicate respective compression processes for the frames, and wherein the method further comprises:

determining available bandwidth on the network (*col. 6, ln. 31-67, determining the flowing conditions, which is related to how fast data are sent*);

selecting a third channel index value from a plurality of channel index values that identify the second time scaling factor, wherein the third channel index indicates a compression process provides highest audio quality at the available bandwidth (*col. 6, ln. 54 to col. 7, ln. 57, a process of adjusting the scale factor to achieve optimum. The process is a continuous*);

requesting from the source a third audio frame that has the third channel index value, which identifies the third audio frame as being time-scaled by the second time-scaling factor (*col. 6, ln. 54 to col. 7, ln. 57, adjust time scale used in the second frame is then again used on the third and subsequent frames*); and

playing the third frame after the second frame to provide a real-time change in the time-scale of the presentation (*referring to figure 1*).

Claim Rejections - 35 USC § 103

17. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

18. Claims 11 and 13 are rejected under 35 U.S.C. 103(a) as being unpatentable over Smyth et al. (US Patent No. 5974380).

19. Regarding claim 11, Smyth et al. further disclose that the apparatus is a standalone device that operates on battery power. However, it would have been obvious to one of ordinary skill in the art at the time of invention that battery is an important source of power used in a variety of devices and systems when other main sources of power are not accessible.

20. Regarding claim 13, Smyth et al. further disclose a method of coding and decoding signal using codebooks (*col. 12, ln. 55 to col. 13 ln. 17*), but fail to specifically disclose that audio frames that are in different channels and have the same frame index represent the same portion of the audio presentation. However, it would have been obvious to one of ordinary skill in the art at the time of invention that if both channels having same frame index, then both frames are the same because each entry in the codebook is associated with only one specific frame index.

21. Claims 5 and 8-9 are rejected under 35 U.S.C. 103(a) as being unpatentable over Smyth et al. (US Patent No. 5974380) in view of Near et al. (US Patent No. 5995091).

22. Regarding claims 5 and 9, Smyth et al. do not disclose that the data structure further comprises a data channel identifying graphics associated with the audio presentation; and a server connected to a network. However, Near et al. teach that the data structure further comprises a data channel identifying

graphics associated with the audio presentation (*col. 7, ln. 25 to col. 8, ln. 12*); and a server connected to a network (*element 702 in figure 7, signal are received from the a remote location sent through the network*).

Since Smyth et al. and Near et al. are analogous art because they are from the same field of endeavors, it would have been obvious to one of ordinary skill in the art at the time of invention to modify Smyth et al. by incorporating the teaching of Near et al. in order to provide timed and coordinated playback of images and/or sounds to multiple users despite differences in playback system speed or configuration.

23. Regarding claim 8, Smyth et al. do not disclose that the data structure further comprises a data channel corresponding to a plurality of bookmarks, wherein each bookmark has index value and identifies graphics, the index value indicating a display time for the graphics relative to playing of the frames of the first or second audio channel. However, Near et al. teach that the data structure further comprises a data channel corresponding to a plurality of bookmarks, wherein each bookmark has index value and identifies graphics, the index value indicating a display time for the graphics relative to playing of the frames of the first or second audio channel (*col. 8, ln. 13 to col. 9, ln. 67*).

Since Smyth et al. and Near et al. are analogous art because they are from the same field of endeavors, it would have been obvious to one of ordinary skill in the art at the time of invention to modify Smyth et al. by incorporating the

teaching of Near et al. in order to present appropriate audio and image files to appropriate subscribers.

24. Claims 20-21 are rejected under 35 U.S.C. 103(a) as being unpatentable over Ware (US Patent No. 5664044) in view of Smyth et al. (US Patent No. 5974380).

25. Regarding claims 20-21, Ware does not disclose that the first frame has a first frame index value that identifies the first portion of the presentation that the first audio frame represents, and the second frame has a second index value that identifies a second portion of the presentation that the first audio frame represents; and the second index value immediately follows the first time index value.

However, Smyth et al. teach that the first frame has a first frame index value that identifies the first portion of the presentation that the first audio frame represents, and the second frame has a second index value that identifies a second portion of the presentation that the first audio frame represents (*col. 33, ln. 40-67 or referring to figure 25*); and the second index value immediately follows the first time index value (*decoding operation is a continuous process*).

Since Ware and Smyth et al. are analogous art because they are from the same field of endeavors, it would have been obvious to one of ordinary skill in the art at the time of invention to modify Ware by incorporating the teaching of Smyth

et al. in order to enable system to correctly decode each frames to enhance system's efficiency.

26. Claims 24-25 are rejected under 35 U.S.C. 103(a) as being unpatentable over Ware (US Patent No. 5664044) in view of Levine et al. (US Patent No. 5886276).

27. Regarding claim 24, Ware discloses a method for playing an audio presentation on a receiver that is connected via a network to a source having a multi-channel data structure representing the audio presentation, the method comprising:

determining available bandwidth on the network (*col. 6, ln. 31-67, determining the flowing conditions, which is related to how fast data are sent*); selecting a first channel of the multi-channel data structure from a plurality of channels that represent the audio presentation after time-scaling by a desired time-scaling factor (*col. 6, ln. 1-30, time-scaled the signal at decoding*); receiving a first frame from the first channel (*col. 6, ln. 1-30, first frame comes first*); and playing the first frame (*referring to figure 1*).

Ware does not disclose that the first channel contains data that is compressed using a compression process that provides highest audio quality at the available bandwidth. However, Levine et al. teach that the first channel contains data that is compressed using a compression process that provides highest audio quality at the available bandwidth (*col. 11, ln. 45 to col. 12, ln. 25*).

Since Ware and Levine et al. are analogous art because they are from the same field of endeavors, it would have been obvious to one of ordinary skill in the art at the time of invention to modify Ware by incorporating the teaching of Levine et al. in order to achieve high transmission rate and good audio quality.

28. Regarding claim 25, Ware further disclose that determining bandwidth available on the network after receiving the first frame (*col. 6, ln. 31-67, determining the flowing conditions, which is related to how fast data are sent*); selecting a second channel of the multi-channel data structure from the plurality of channels that represent the audio presentation after time-scaling by the desired time-scaling factor (*col. 6, ln. 1-30, time-scaled the signal at decoding and the system is a multi-channel system*); receiving a second frame from the second channel (*col. 6, ln. 1-30, first frame comes first*); and playing the second frame after playing the first frame (*col. 6, ln. 1-30, first frame comes first*).

Ware does not disclose that the second channel contains data that is compressed using a second compression process that provides highest audio quality at the bandwidth available after receiving the first frame. However, Levine et al. teach that the second channel contains data that is compressed using a second compression process that provides highest audio quality at the bandwidth available after receiving the first frame (*col. 11, ln. 45 to col. 12, ln. 25*).

Since Ware and Levine et al. are analogous art because they are from the same field of endeavors, it would have been obvious to one of ordinary skill in the

art at the time of invention to modify Ware by incorporating the teaching of Levine et al. in order to achieve high transmission rate and good audio quality.

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Huyen Vo whose telephone number is 703-305-8665. The examiner can normally be reached on M-F, 9-5:30.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Doris To can be reached on 703-305-4827. The fax phone number for the organization where this application or proceeding is assigned is 703-872-9306.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

Examiner Huyen X. Vo

July 12, 2004

W. R. YOUNG
PRIMARY EXAMINER